



ISSN (Print) : 2320 – 3765
ISSN (Online): 2278 – 8875

International Journal of Advanced Research in Electrical, Electronics and Instrumentation Engineering

(An ISO 3297: 2007 Certified Organization)

Vol. 3, Issue 7, July 2014

[DOI: 10.15662/ijareeie.2014.0307066](https://doi.org/10.15662/ijareeie.2014.0307066)

A New Approach of Speech Compression by Using DWT & DCT

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ABSTRACT: Compared to most digital data types, with the exception of digital audio, the data rates associated with uncompressed digital audio are substantial. Digital audio compression enables more efficient storage and transmission of audio data. The many forms of audio compression techniques offer a range of encoder and decoder complexity, compressed audio quality, and differing amounts of data compression. In this paper a new algorithm for speech signals compression using wavelet transform technique with Discrete Cosine Transform (DCT) technique. The performance of the implemented algorithm is evaluated based on Signal to Noise Ratio (SNR), Root Mean Square Error (MSE) and compression ratio tested on speech signals. In this paper a Wavelet & cosine hybrid model, based speech coder is implemented in software using Matlab.

KEYWORDS: DWT, DCT, Speech compression & decompression, MSE, SNR.

I. INTRODUCTION

Speech/Audio coding has been and still is a major issue in the area of digital speech processing. Speech coding is the act of transforming the speech signal at hand, to a more compact form, which can then be transmitted with a considerably smaller memory. The motivation behind this is the fact that access to unlimited amount of bandwidth is not possible. Therefore, there is a need to code and compress speech signals. Speech compression system focuses on reducing the amount of redundant data while preserving the integrity of signals. The different transformation of speech signals to the time frequency and time-scale domains for the purpose of compression aim at representing them with the minimum number of coding parameters. Digital audio compression allows the efficient storage and transmission of audio data. The various audio compression techniques offer different levels of complexity, compressed audio quality, and amount of data compression. Compression systems are designed to eliminate this redundancy. Speech coding is a lossy type of coding, which means that the output signal does not exactly sound like the input. The input and the output signal could be distinguished to be different. Coding of audio however, is a different kind of problem than speech coding.

a) Wavelet Compression

Different techniques were implemented each having its advantages and disadvantages. In general, they can be classified into two types, lossy and lossless. The original data will be totally restored without any modifications. Lossy compression does not completely retain the original signal; consequently some of the information is lost. For speech signals, this loss is acceptable since we are interested only in recognizing the signal. Wavelets compression technique is considered to be lossy where the reconstructed signal is not an exact match of the original signal. When applying the Discrete Wavelet Transform (DWT) to a given speech signal many coefficients of small values (depending on level we choose) are thus considered insignificant but ignoring them may cause more redundancy in recovered in sound signal.

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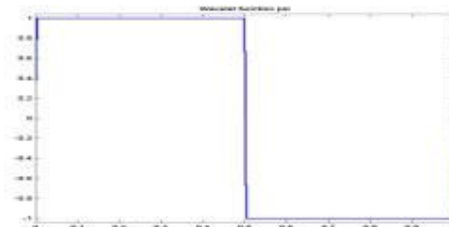
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b) Wavelet Families

It is very important to briefly introduce wavelets' families, because they are the main tools. For anyone who wants to study wavelet theory must be familiar with these families, in order to get the best performance of his work. We can't say that one type of wavelet is better than another, because every type has its own applications. So, a wavelet family may be good for one application, but not for another, this depends on the nature of the application. We will introduce, for each wavelet family, the scaling function $\Phi(t)$ in Eq. (1), the wavelet function $\Psi(t)$ in Eq. (2) and its filters' values (for a filter bank) in both of time-domain and frequency-domain representations.

$$\Phi(t) = 2 \sum_{k=0}^N h(k) \Phi(2t - 1) \quad (1)$$

$$\Psi(t) = 2 \sum_{k=0}^N h(k) \Phi(2t - 1) \quad (2)$$



Haar wavelet

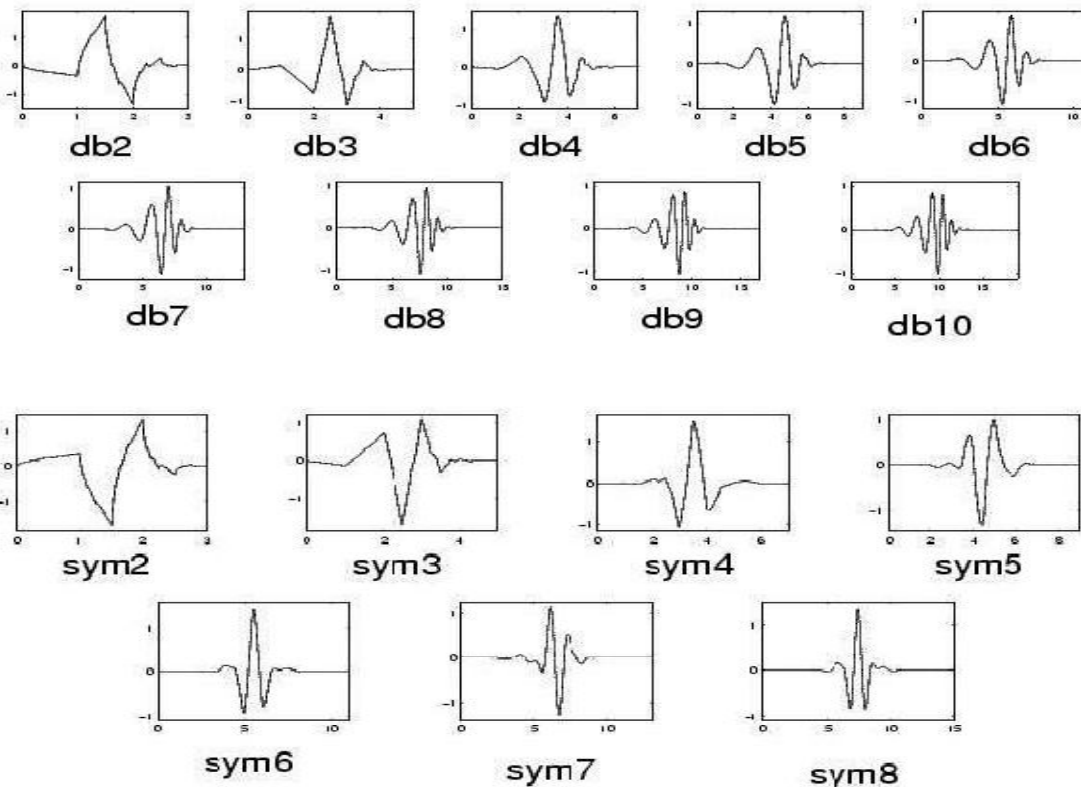


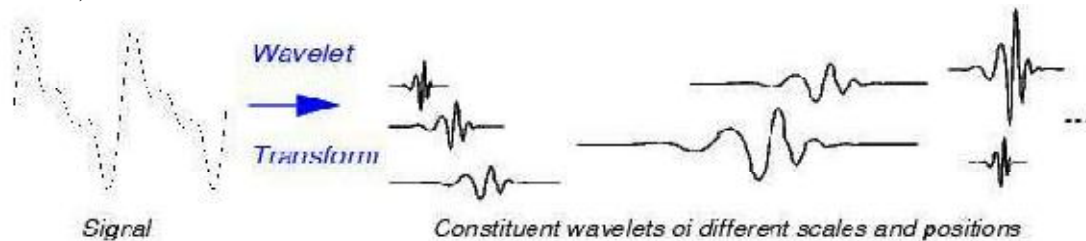
Fig1: some wavelet function

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Wavelet analysis is the breaking up of a signal into shifted and scaled versions of the original (or mother) wavelet.



c) Discrete Cosine Transform

Discrete Cosine Transform can be used for speech compression because of high correlation in adjacent coefficient. We can reconstruct a sequence very accurately from very few DCT coefficients. This property of DCT helps in effective reduction of data.

II. IMPLEMENTATION

In this paper hybrid model of compression and decompression by using DWT and DCT technique is used for the recorded signal. A different approach is being used for better compressed signal as well as less redundant signal is being recovered.

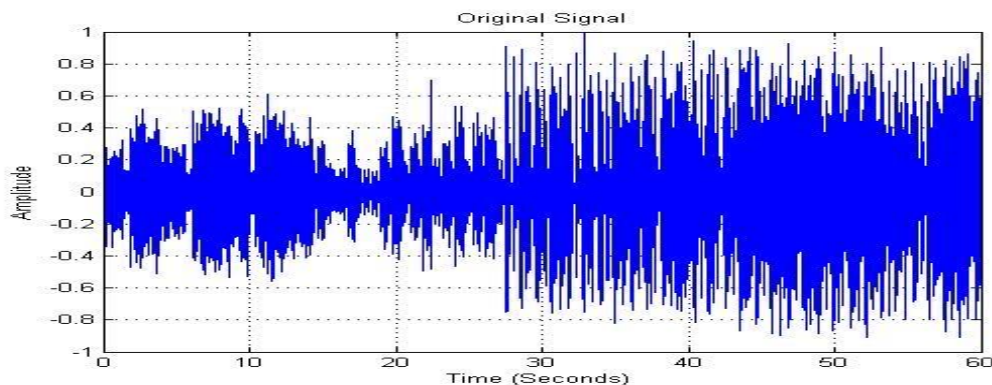


Fig 1: Original sound signal

Set the compression and decompression flag at zero level. Then set the compression ratio i.e. up to what compression level is required. By using the frame size of the signal we can estimate the size (length) of the signal to be compressed. After the transformation of the signal, we arrange the transformed coefficient so that lower value of frequency of the signal is not eliminated. Whereas we are move the unnecessary zeros to other location. This provides a great help in compression with better result of MSE and SNR. In the above figure the signal which is used to compress using DWT then by DCT with different compression ratio generating the compressed signal. After the compression by removing the distortion or unnecessary coefficient the signal is reconstructed to the original form by taking inverse transform after adding the zeros which were generated in DWT & DCT. In method of compression the signal doesn't lose any of its information.

In this paper compression of the speech signal is compared among the filters of the wavelet family such as haar, db4, sym, coif2. Now take the frame size of the signal to be 10 sec to analysis the different compression ratio in order to attain compressed signal with better result. Comparison is on the basis of mean square error (MSE) and signal to noise ratio (SNR).



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Table 1: compression ratio 2

FILTER	MEAN SQUARE ERROR	SIGNAL TO NOISE RATIO
Haar	5.386e-05	23.1941
db4	4.1441e-05	24.3325
Sym	4.3722e-05	24.0998
Coif2	3.936e-05	24.5265

Table 2: compression ratio 4

FILTER	MEAN SQUARE ERROR	SIGNAL TO NOISE RATIO
Haar	0.00038263	14.679
Db4	0.00026461	16.2808
Sym	0.00029704	15.7786
Coif2	0.00027087	16.1792

Table 3: compression ratio 6

FILTER	MEAN SQUARE ERROR	SIGNAL TO NOISE RATIO
Haar	0.00075946	11.7018
db4	0.00054662	13.1299
Sym	0.00060383	12.6977
Coif	0.00054669	13.1294

III. CONCLUSION

Speech compression is used for transmission and storage. The speech compression is achieved by representing each sample of digitized data by lesser number of bits this paper shows the key advantageous features of different Wavelet filters in the field of speech Signal processing. It is found that the different wavelet filters significantly improves the reconstruction or fidelity assessments of the compressed speech signal. The SNR and MSE of signal show the high efficiency of the compression it is concluded that it can be very effectively used for the speech signal compression.

IV. RESULT

Speech compression is achieved by using HAAR, DB4, SYM, COIF filter of DWT and DCT transform. In table 1 to 3 shows the MSE, SNR of the given speech signal of frame size as 10sec of the recorded speech signal. The compressed signal with compression ratio 2 of different filter of wavelet transform gives us the better result in terms of reconstructed speech signal.

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